APPLICATION FOR PATENT

Inventor: Eran Shpak

Title: METHOD AND SYSTEM FOR VOICE-OVER-IP COMMUNICATION

FIELD AND BACKGROUND OF THE INVENTION

The present invention relates to voice-over-IP communication and, more particularly, to a method and system of point-to-multipoint voice-over-IP communication.

The Ethernet protocol is widely used in point-to-multipoint communication in local area networks (LANs). Figure 1 shows a typical Ethernet packet **10**. More specifically, packet **10** is a "RTP over Ethernet" packet. Packet **10** includes a header **12** followed by a payload **14**. Header **12** includes 14 Ethernet header bytes followed by 40 IP/UDP/RTP bytes. The number of bytes in payload **14** is application dependent. For voice communication, payload **14** typically is a G.729 payload of 20 bytes. All the bytes of packet **10** are eight bits long

Packet **10** would be an inefficient vehicle for voice-over-IP in other point-to-multipoint systems, particularly in wireless systems, for two reasons. First, in wireless systems, bandwidth is at a premium. The high ratio of header bytes to payload bytes in packet **10** would make inefficient use the bandwidth of a wireless system. Second, at 74 bytes total length, packet **10** is relatively short. Voice-over-IP using packet **10** would entail transmitting a relatively large number of relatively short packets. Wireless systems are most efficient when a relatively small number of relatively long packets are transmitted.

There is thus a widely recognized need for, and it would be highly advantageous to have, a method of wireless voice-over-IP communication that makes more efficient use of the available bandwidth than presently known methods.

In point-to-point packet communication, for example, in data and voice communication over the Internet, it is common to increase the efficiency of the communication by header compression. Once a communication session is established, the portion of the packet header that will remain constant during the course of the session is replaced by a shorter (typically two bytes long) token. A transmitting party replaces the constant part of the header with the token, and the receiving party expands the token into the constant part of the header. It should be noted that header compression is not used in Ethernet LAN voice-over-IP.

A NOTE ON NOMENCLATURE

Under the IEEE standard, the entities referred to herein as "packets" are called "frames", and the entities referred to herein as "8-bit bytes" are called "octets".

SUMMARY OF THE INVENTION

According to the present invention there is provided a method of transmitting a plurality of voice communications from respective end points to an access point, including the steps of: (a) providing a point-to-multipoint network operative to send packets from the end points to the access point; (b) for each end point: (i) negotiating a respective alias with the access point, and (ii) configuring the respective voice communication as a voice payload; (c) concatenating a single superpacket header with the aliases and with the voice payloads to form a superpacket; and (d) sending the superpacket to the access point via the point-to-multipoint network.

Preferably, the point-to-point network is configured according to OSI layer 2.

Preferably, the access point receives the superpacket and unbundles the superpacket into a plurality of received packets, with each received packet corresponding to a respective voice packet and with each received packet including a header configured according to the respective alias.

Preferably, the voice payloads are G.729 payloads.

Preferably, the superpacket header is an Ethernet header.

Preferably, the aliases and the voice packets are interleaved within the superpacket.

Preferably, the superpacket header includes a type field that indicates that the superpacket header is followed by the aliases and by the voice packets.

Preferably, each alias includes a respective station ID, and the negotiating of the aliases includes negotiating the respective station IDs.

Preferably, the voice packets are synchronized prior to being concatenated to form the superpacket.

Preferably, the negotiating and the concatenating are effected only by providing, in the point-to-multipoint network, a voice-over-IP gateway operative to effect the negotiating and the concatenating, and then effecting the negotiating and the concatenating using the voice-over-IP gateway.

According to the present invention there is further provided a system for transmitting a plurality of voice packets from respective end points to an access point, including: (a) a voice-over-IP gateway for: (i) negotiating, with the access point, a respective alias for each end point, (ii) receiving, from each end point, a respective voice payload, and (iii) concatenating a single superpacket header with the aliases

and with the voice payloads to form a superpacket; and (b) a mechanism for transmitting the superpacket to the access point.

Preferably, the mechanism includes either a wireless point-to-multipoint network or a cable TV point-to-multipoint network.

Preferably, the mechanism is configured according to OSI layer 2.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention is herein described, by way of example only, with reference to the accompanying drawings, wherein:

FIG. 1 (prior art) shows a RTP over Ethernet packet ;

FIG. 2 shows a system of the present invention;

FIG. 3 shows a superpacket of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

The present invention is of a method and system that can be used for efficient voice-over-IP communication over a wireless point to multipoint network. The present invention also is suitable for use in other point to multipoint networks, for example, cable TV networks.

The principles and operation of point-to-multipoint voice-over-IP according to the present invention may be better understood with reference to the drawings and the accompanying description.

Referring again to the drawings, Figure 2 illustrates a system **20** of the present invention. Three end points (IP telephones) **22** are connected by respective twisted wire pairs **28** to a voice-over IP gateway **24**, which in turn is connected via a LAN **30** to a mechanism **26** that provides wireless RF communication to an access point **32**.

Access point **32** is in turn connected to a public switched telephone network (PSTN) **34** via a second LAN **36**. In the absence of gateway **24**, end points **22** would be connected directly to mechanism **26** via LAN **30**, and system **20** would be a prior art point-to-multipoint network that exchanged voice-over-IP packets between end points **22** and PSTN **34** without header compression and packet bundling as described below. In particular, this prior art network would be configured according to OSI layer 2. Such networks also are referred to in the art as "OSI layer 2 bridges", or equivalently as "MAC (medium access control) layer bridges". A particular feature of the present invention is that gateway **24** can be inserted into a prior art OSI layer 2 bridge, as shown, *with no other change to the existing infrastructure. In particular, no change needs to be made to the underlying OSI levels 1 (wireless PHY) and 2 (MAC) layers.*

End points **22** transmit packets similar to packet **10** to gateway **24**. In particular, the packets transmitted by end points **22** include G.729 voice payloads, each with 20 8-bit bytes. Gateway **24** performs header compression and bundling on these packets to create a "superpacket" that is transmitted to access point **32** via mechanism **26**. The header compression performed by gateway **24** is similar to the header compression performed in prior art point-to-point voice-over-IP systems, except that because several end points **22** may be communicating with access point **32** simultaneously, each time a specific end point **22** establishes a communication session with access point **32**, that end point **22** and access point **32** negotiate a unique 2-byte "circuit description" alias to use for that communication session. This circuit description alias includes a 6-bit station ID that identifies that specific end point **22**.

Figure 3 shows a superpacket **40** created by gateway **24** for transmission to access point **32**, in the course of three simultaneous communication sessions by end

points **22** with access point **32**. It is assumed that gateway **24** has received: from end point **22a**, a respective voice-over-IP packet including a voice payload of 20 8-bit bytes; from end point **22b**, a respective voice-over-IP packet including a voice payload of 20 8-bit bytes; and from end point **22c**, a respective voice-over-IP packet including a voice payload of 20 8-bit bytes. Superpacket **40** includes 74 8-bit bytes in the following order: an Ethernet-like header field **42** of 12 bytes, a type field 44 of two bytes, a circuit description field **46a** of two bytes, a payload field **48a** of 20 bytes, a circuit description field **46b** of two bytes, a payload field **48b** of 20 bytes, a circuit description field **46c** of two bytes and a payload field **48c** of 20 bytes. Header field **42** and type field **44** together constitute the header of superpacket **40**. Header field **42** is similar to the first 12 bytes of header **12**. Type field **44** contains a preselected code that indicates to access point **34** that superpacket **40** is formatted according to the present invention. The inclusion of type field **44** in superpacket **40** allows the use of other preselected codes to indicate that superpacket **40** is formatted according to prior art formats, thereby enabling system **40** to transparently transmit either superpackets **40** of the present invention or prior art packets such as packet **10**. Circuit description fields **46a**, **46b** and **46c** contain, respectively, the circuit description aliases that have been negotiated between access point **32** and end points **22a**, **22b** and **22c**. Payload fields **48a**, **48b** and **48c** contain, respectively, the voice payloads received from end points **22a**, **22b** and **22c**.

Because the transmission of the separate packets from end points **22** to gateway **24** is not synchronized, gateway **24** also synchronizes the received packets before assembling superpacket **40**.

Gateway **24** sends superpacket **40** to access point **32** via mechanism **26**. By inspecting the contents of type field **44**, access point **32** determines that the packet it

has received is a superpacket **40** of the present invention Access point **32** then unbundles superpacket **40** into three packets **10**, with payload **14** of each packet **10** being the voice payload carried by a respective payload field **48a**, **48b** or **48c** of superpacket **40** and with header **12** of each packet **10** being constructed in accordance with the contents of the respective circuit description field **46a**, **46b** or **46c**. Access point **32** then sends the three packets **10** to PSTN **34** via LAN **36**.

If the contents of type field **44** indicate that a received packet is a prior art packet rather than a superpacket **40** of the present invention, then gateway **24** sends the packet directly to PSTN **34** via LAN **36**.

The presence of three circuit descriptor fields **46** and three payload fields **48** in superpacket **40** is only exemplary. Superpacket **40** can be configured with any convenient number of circuit descriptor fields **46** and associated payload fields **48**.

As an example of the more efficient bandwidth use of the present invention, consider a system **20** configured according to the prior art (i.e., without gateway **24**) and sending three packets **10** (one packet from each of end points **22**) to PSTN **34** every 20 milliseconds (150 packets per second) via mechanism **26** and access point **32**, hence with a link latency of 20 milliseconds. Each packet **10** is 74 8-bit bytes long, so the bandwidth per channel of mechanism **26** is 74*150*8/3=29,600 bps. Contrast this with system **20** configured according to the present invention with gateway **24** and sending one superpacket **40** every 20 milliseconds to access point **32** via mechanism **26**. Superpacket **40** is 80 8-bit bytes long, so the bandwidth per channel of mechanism **26** is 80*50*8/3=10,667 bps, which is almost a threefold improvement over the prior art, with no increase in link latency.

While the invention has been described with respect to a limited number of embodiments, it will be appreciated that many variations, modifications and other applications of the invention may be made.